Integrating Cisco IP Telephony and IP/VC Videoconferencing Networks

An AVVID Application Note

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Table of contents

Introduction

Today, many customers are deploying large-scale IP telephony networks using Cisco AVVID solutions such as CallManager and IP telephones. Many of these customers also are deploying H.323 videoconferencing networks using Cisco's H.323 videoconferencing solutions such as the IP/VC 3500 series MCU's and gateways, as well as the MCM (Multimedia Conference Manager) gatekeeper and proxy. This paper will discuss how to merge these technologies together.

Let's consider a business scenario to help you understand how this techology integration can be of benefit.

A Business Example

There is a meeting at 10:00AM between three locations, where the Engineering team is discussing the next version of a product. These three sites are connected via an IP network, and use H.323 videoconferencing technology to conduct a multipoint conference. A Cisco IP/VC MCU is used to bridge these three sites together, so that everyone can hear and see one another.

However one of the participants is traveling, and wants to be able to dial in from his or her cellphone, since they won't be able to attend in person. Similarly, one of the participants is in a remote office that does not have a videoconferencing station, so they would like to dial in from the Cisco 7960 IP phone on their desk.

How do you bring these three technologies (POTS or cellular phones, IP phones and H.323 videoconferencing) together in one conference? Historically, people would probably use an audio conferencing service such as Latitudes' Meeting Place for the audio, and the H.323 MCU for the videoconference. In other words, even the sites that have video capabilities would also have to dial in to the audio conferencing bridge, and mute the audio on the videoconferencing device. This results in twice the cost per conference! Not to mention a very awkward user experience since they will be looking at the other sites on the videoconferencing display, but listening to them through a telephone. Lip-sync in this situation is not preserved, since the audio path in 99 percent of the cases will have less end-to-end latency than the videoconferencing path, and so they will hear the people's voices coming through on the phone before they see their lips moving on the videoconferencing display.

Goals of This Paper

The primary goal of this paper is to demonstrate how to configure your IP telephony and H.323 videoconferencing networks so that users can seamlessly call one another. The network elements should make the call signaling differences between IP telephony and H.323 transparent. In addition, attaining this goal should also allow for seamless interconnectivity to the PSTN for both audio and videoconferencing users. The secondary goal of this paper is to specifically point out how the IP/VC 3500 series MCU's can be used as an integrated conference bridge for both audio and video participants.

There are many aspects to such a converged solution, such as a unified dial plan, QoS (Quality of Service) and many other aspects. This paper will not deal in detail with these items, but will focus solely on how to configure the necessary call routing between these three technologies. QoS is a subject all by itself, and several excellent whitepapers, design guides and presentations already exist to deal with this topic. Likewise, the topic of building large-scale unified dial plans could also consume an entire paper by itself. For the purposes of this paper, we will deal with very simplistic dial plans, which will provide enough information to get the point across. You can then apply this information when creating larger deployment scenarios.

In addition, the ultimate goal of this document is to explain how this can be done *today*, and briefly outline what will be available in the near future. Therefore, the reader will know what *can* and *cannot* be done *today*, giving customers a realistic understanding of the state of the capabilities that can be achieved with such integration.

This paper, then, will deal with the following 5 topics:

1. Explain how the Cisco IP/VC 3510 and 3540 MCU's can be used to support both voice and video endpoints, seamlessly joining them together in one conference call. This subject will also include a brief overview of the current limitations of the 3500 series MCU's, and what future software releases will do to alleviate them, thus providing you with a realistic view of how the 3500 series MCU's should be positioned *today* and in the future.
2. Explain how CallManager can be configured to interface to an H.323 videoconferencing network, and provide routing of calls from an IP phone to an H.323 videoconferencing device.
3. Explain how the Cisco MCM gatekeeper can be configured to interface with CallManager, and provide routing of calls from an H.323 videoconferencing device to an IP phone.
4. Explain how PSTN users (i.e. POTS phones, cellular phones and H.320 videoconferencing devices) can use gateways to access both H.323 videoconferencing devices and IP phones.
5. Explain how the Cisco MCM gatekeeper can be configured to route calls from an H.323 videoconferencing device to PSTN users (i.e. POTS phones, cellular phones or H.320 videoconferencing devices).

Reference Documentation

Where appropriate, this paper refers to existing procedures and information in the following Cisco documents:

Cisco CallManager Administration Guide

Cisco Multimedia Conference Manager Configuration Guide

Cisco IP/VC 3500 Series Configuration Guides

Cisco Videoconferencing Design Guide

Cisco IP Telephony Design Guide

Cisco QoS Design Guide

Routing Gateway and MCU Service Prefixes on the Cisco MCM Gatekeeper – Whitepaper

Configuring Services for the Cisco IP/VC 3510 and 3540 MCU's – Whitepaper

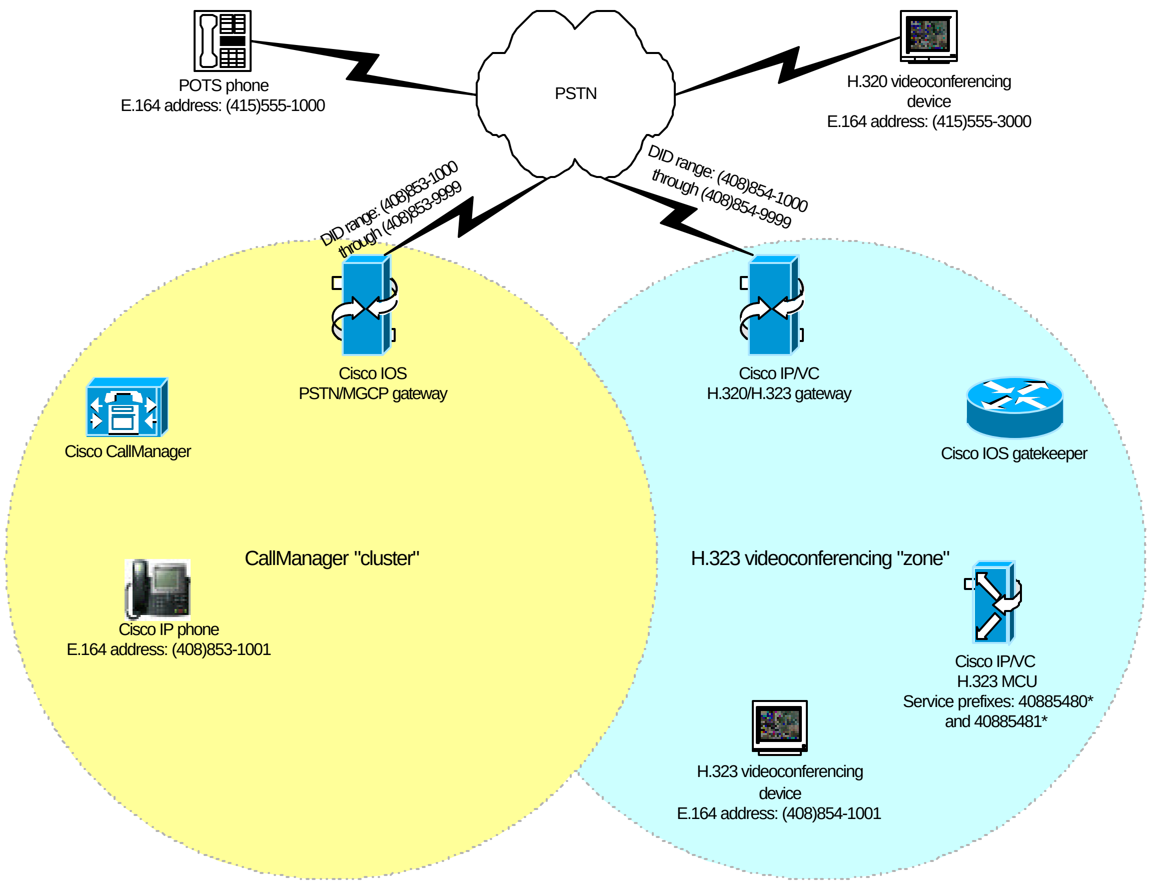
Call Survivability in CallManager 3.0(x) - Whitepaper

Cisco CallManager-IP/VC Interoperability – EDCS Document# ENG-105282

Note: The procedures in this paper assume familiarity with Cisco's IP telephony solutions, IP/VC videoconferencing solutions, IOS Software, Multimedia Conference Manager, and H.323. For instructions on configuring these devices and services, please refer to the appropriate users guide.

Diagram example of a converged voice/video network

Figure 1: Logical network topology example



The network depicted in figure 1 is the model that we will use for the examples given in this paper. It illustrates an IP telephony network, managed by a Cisco CallManager and containing devices that use the Skinny Station Protocol; a PSTN network, consisting of POTS phones, cellular phones, and H.320 devices; and an H.323 videoconferencing network managed by a Cisco MCM gatekeeper and containing devices that use the H.323 protocol.

Although the signaling protocols used in each of these networks differs, any-to-any connectivity can be achieved by following the recommendations presented in this paper. To achieve this, a bi-directional call signaling connection between the CallManager cluster and the H.323 videoconferencing network must be established. In addition, PSTN and H.320 ISDN connectivity can be achieved through a variety of gateway options, including PSTN/Skinny gateways, IOS-based PSTN/H.323 or PSTN/MGCP gateways, and IP/VC H.320/H.323 gateways.

DN (Directory Number) ranges are assigned to both the CallManager cluster and the H.323 zone. In addition, DN's are also assigned by the local telephone company to the gateways PSTN interfaces, whether they are FXO, FXS, E&M, T-1 CAS, ISDN BRI or PRI.

Finally, E.164 addresses are assigned to the individual end stations, whether they be IP phones, videoconferencing terminals, or videoconferencing MCU's (Multipoint Control Units).

This dial plan allows any device to reach any other device by dialing its E.164 address or equivalent DN. In the case of the MCU, calls must be made to it by dialing its H.323 zone prefix and service prefix, along with the conference ID of the conference they want to join. The CallManager and the H.323 gatekeeper both have route patterns established to route these calls to the appropriate destination. The IOS H.323 gateways would also have the equivalent "dial peers" configured.

After examining the network illustrated in Figure 1, you may have several questions, such as,

- "How do you configure the CallManager to route calls to the H.323 videoconferencing network?"
- "How do you configure the H.323 gatekeeper to route calls to the CallManager cluster?"
- "Does the Cisco IP/VC MCU support audio-only participants, so that I can have IP phones and POTS phones call into an MCU conference, along with my video terminals?"

Let's start by looking at the capabilities of the Cisco IP/VC 3500 series MCU's to support audio-only participants.

Capabilities of the Cisco IP/VC 3500 series MCU's

The Cisco IP/VC 3510 and 3540 MCU's are H.323 devices which support a variety of call speeds, ranging from 64kbps to 2Mbps, and support both audio and video algorithms. It has been marketed by Cisco as a videoconferencing solution, and its ability to support audio only participants has often been overlooked. One reason for this is because the 3510 MCU had very limited capacity, and was not specifically tailored to supporting AVVID voice deployments.

However, due to the increased capacity of the 3540 (sixes times the capacity of the 3510), the AVVID marketing team feels that the capabilities of the MCU to serve as an audio bridge, as well as a video bridge, should be re-examined and highlighted.

Both the 3510 and 3540 MCU models support G.711 audio, as well as H.261 and H.263 video. They are H.323v2 compliant, and register their E.164 prefixes with a gatekeeper. They also support two modes of operation: "Ad Hoc" and "Scheduled Only". We will deal with each of these items in more detail below.

When a terminal calls into the MCU, they perform the capabilities exchange process per the H.245 standard. Therefore, it is capable of supporting a mix of voice/video and voice-only participants. When the terminal dials in, it notifies the MCU what it is capable of. In the case of an IP phone or a POTS phone, they inform the MCU that they are capable of audio only (video is not present in their capabilities message). Therefore, the MCU will allow them to join the conference, but will only open a logical channel for audio. The only real difference between this scenario and a video terminal dialing in is that the video terminal informs the MCU that it is capable of both audio and video, and the MCU will open logical channels for both media types. All of the above devices use RTP to transport the audio packets. In the case of a POTS or cellular phone, the analog audio is converted to RTP format by the gateway.

So what this means from a user perspective is if you had four video terminals dialed in to an MCU conference, and someone dialed in from a telephone, the people on the video terminals would hear the telephone user, but would not see them (obviously). Likewise, the telephone user will hear everyone else, but will not participate in the video session.

The Cisco IP/VC MCU's mix the audio of the four most dominant sites. In other words, the terminals that are sending the four strongest levels of audio at a given moment in time will be mixed together, and that combined audio stream is what everyone else in the call will hear. This behavior is the norm for conference bridges. If you have twenty stations dialed into the conference, the four loudest sites will be the ones that are heard by everyone else. This mixing happens dynamically throughout the duration of the call.

Both IP/VC MCU models offer an administrative web-interface to control conferences being hosted on that MCU. The web-interface on the 3540 is significantly better than its older 3510 counterpart, in that the MCU is completely controlled by a web interface, which offers three level of access: Administrator, Conference Manager and User. Only people with administrative rights can access the administration interface. People with Conference Manager privileges can access the MCU's conference management interface, and add, remove or control conference participants. People with User privileges can view the conference, see who is connected, and invite other participants into the call, but cannot remove participants or take control of the conference in any other way.

Using this web-interface, if three sites were connected in a videoconference, and they wanted to add a telephone user to the conference, someone with Conference Manager or User privileges could access the web-interface and "invite" the telephone user to the call, by entering the appropriate telephone number/ dial-string, and pressing the "invite" button.

Note: For more information on using the web-interface on either of IP/VC 3500 series MCU's, please refer to the users guides referenced at the beginning of this document.

In addition, the MCU can be configured to operate in “Ad Hoc” mode, where anyone can start a conference on the fly by simply dialing in to the MCU and then inviting others to the conference. Alternatively, the MCU can be configured for “Scheduled Only” mode. In this mode, users can access a scheduling server to setup a conference in advance for sometime in the future. When that time comes, the MCU can either dial out to all the participants, or can wait for them to dial in.

Finally, the MCU offers a feature called “Dynamic Expansion”. This feature allows the Administrator to create a conference service for a limited amount of participants, but if more people decide to join than was originally planned, the MCU will allow them in only if CPU resources are available. In other words, one could create a conference service for four participants. But if a fifth person dialed in, the MCU could allow that person in by dynamically expanding the size of the conference, if there are enough CPU resources available at the time of the request, and doing so would not jeopardize the resources of any other conferences.

Limitations of the IP/VC 3500 series MCU’s

There are some features that the IP/VC 3500 series MCU’s are missing for it to be used as a high-end conferencing solution for voice-only applications. However, there are also some very succinct benefits, mainly circling around its ability to bring audio and video devices together in a combined conference. This integration is another key step in Cisco’s momentum around the AVVID architecture.

Here is a list of items that we feel are missing from the MCU to make it a feature-rich conferencing solution for voice:

1. G.711 only

   In order to compete in the voice arena, other compression schemes are necessary, such as G.723 and G.729. However, we believe that this should not be a barrier to deployment, since there are other mechanisms available to bring various compression schemes together in one conference, such as the MTP (Media Termination Point) resources available in CallManager and on the Catalyst 4k and 6k line cards.

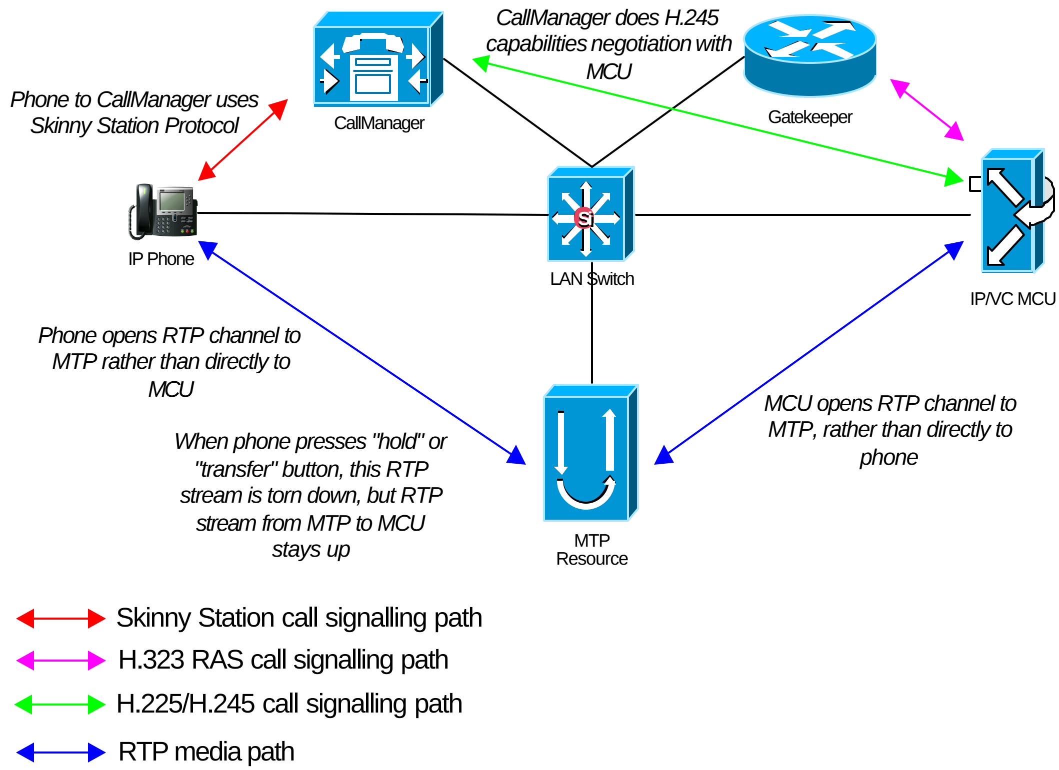
   However, later this year a daughtercard module will be available as a field upgrade for the IP/VC 3540 MCU to provide audio transcoding. It is planned to support G.711, G.723, G.729 and G.722. When this module becomes available, it will merely serve to offer customers an additional option, and minimize the requirement for additional MTP resources elsewhere.

2. Lack of support for H.245 “Empty Capabilities Set”

   The empty capabilities set message allows telephones to place callers on hold, transfer the call to another destination, etc. These features are often collectively referred to as “supplementary services”. The MCU is incapable of supporting the empty capabilities set message. This feature is planned for a future release of the 3540 MCU later this year. Support for this may also be available in some future release of the 3510 MCU.

   What this means is that for the time being, the MCU must be “front-ended” with MTP resources if these supplementary services are desired. In other words, the RTP media stream for a call from an IP phone to an IP/VC MCU must be terminated by a MTP resource, and then a second call leg will be established from the MTP to the MCU. The MTP is capable of handling the hold and transfer features on behalf of the MCU. Below is an example of such a scenario.

Figure 2: Using an MTP to provide supplementary services

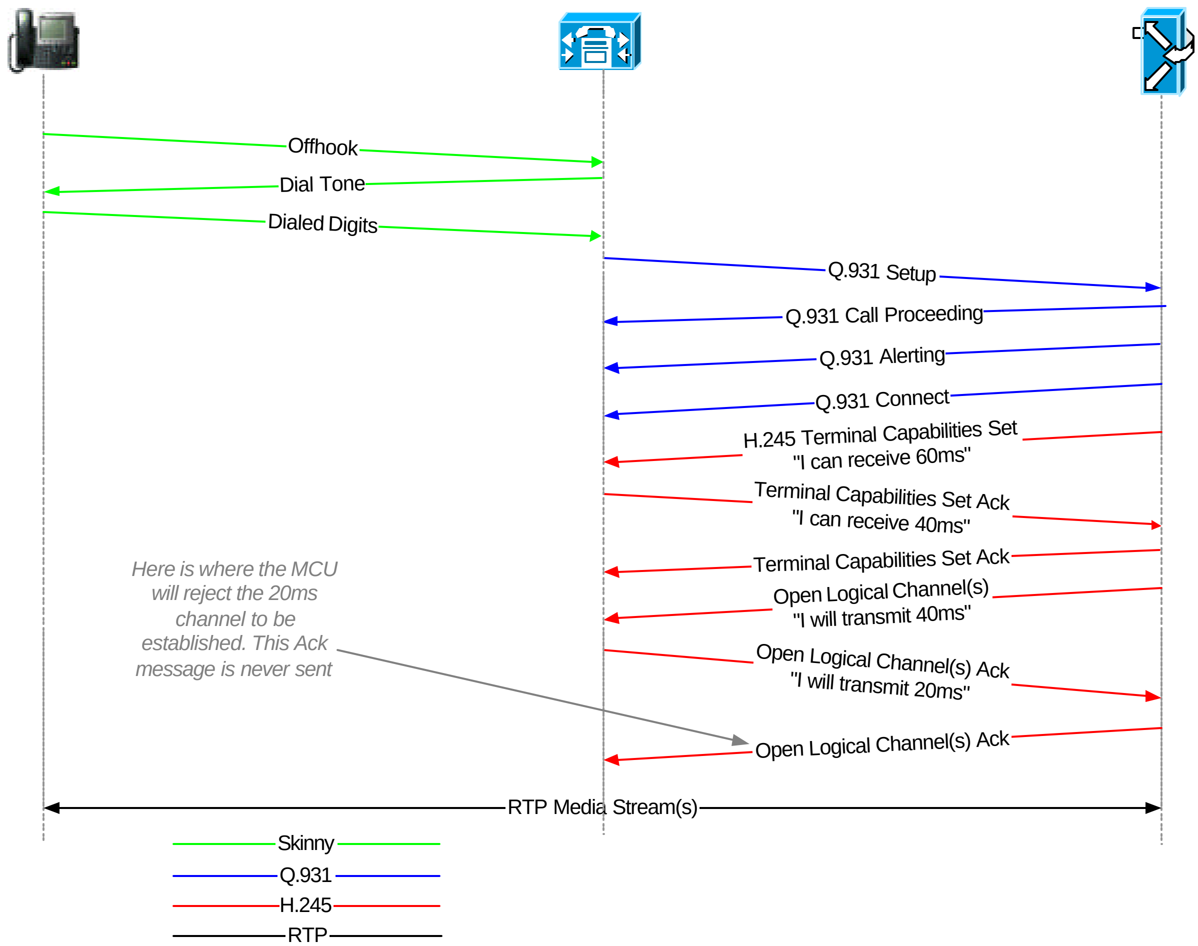


The benefit of using an MTP for each IP phone that participates in the MCU conference is that the phone can put the call on hold, or transfer the call to another telephone. The drawback is that you will need to use an MTP resource for each IP phone that joins the conference. Again, a future release is planned to add support for the empty capabilities set in the MCU, negating the need for MTP's to provide these supplementary services.

3. Lack of support for 20ms packet sizes

The IP/VC 3500 series MCU's support a range from 30ms – 60ms packet sizes. It is capable of receiving as low as 30ms, and will send the largest packet size advertised by the terminal in its H.245 terminal capabilities message. In the case of IP phones, the H.245 terminal capabilities message advertises that the phone is capable of receiving up to 40ms packets. Therefore, when the logical channels are opened In both directions, the MCU will send at 40ms, but will receive at whatever speed the IP phone is configured to send at. This speed is configurable in CallManager, but the default is 20ms. Therefore, if left at its default value of 20ms, the MCU will reject the opening of the logical channel from the phone, since it is only capable of receiving as low as 30ms. You must therefore configure the phones to send at 30ms instead of 20ms. We will provide those configuration details later in this paper. Figure 3 below illustrates the H.245 negotiation process described abve.

Figure 3: H.245 capabilities exchange between CallManager and the IP/VC 3500 series MCU's



This limitation means that you must change the configuration of the phones to use 30ms instead of 20ms, which could have an effect on the rest of your IP telephony network, which often is carefully tuned. Network designers should consider the *possible* larger ramifications to the network before making this adjustment.

Like the above bullet points, support for 20ms is planned for a future release of the 3540 MCU, due out later this year. It is not currently planned for the 3510 MCU, since processing 20ms requires additional CPU horsepower that the 3510 model does not have.

4. Lack of audible entry and exit tones

This is probably the biggest barriers to deployment if the IP/VC 3500 series MCU's are being compared against other high-end voice bridges. There are currently no entry or exit tones, announcements or any other mechanism for letting the people in a conference know that someone else just joined, or that someone left the conference. When people connect to the conference, they are simply "dumped" straight in. This presents a bit of a security concern, because people could potentially "eavesdrop" on the conversation without anyone knowing they are there. Of course, they would have to know the exact number of the conference, and when it is taking place, so there is still a natural amount of "security by obscurity" so to speak.

Entry and exit tones are planned for a future release of the 3540 MCU due later this year. It is unlikely that this will ever be added to the 3510 model.

As you can see from the above points, the 3500 series MCU's should *not* be positioned as a complete replacement for other audio conferencing services, which typically provide all of these features and more. However, we believe that its ability to support audio-only participants is an exciting breakthrough and another step towards convergence, and should be positioned as such. For customers who are deciding on purchasing a conferencing bridge specifically for voice, the 3500 series MCU's are probably not yet ready to be a strong contendor. However, if the customer is simply trying to augment there existing voice conferencing services, and provide for integrated voice and video conferences, the 3500 series, and in particular, the 3540 model, can be an effective tool.

In addition, due to the need for MTP resources, and the cost of the 3500 series MCU's, you should continue to position the CallManager software conferencing resources, or the Catalyst line cards for small, ad hoc conferences, where the user just wants to push the "conference" button on their IP phone, and bridge a third party onto the call. The 3500 series MCU's are more appropriately positioned as an integrated voice/video bridge for meet-me style conferences.

Configuring CallManager

There are three methods for configuring CallManager to route calls to an H.323 videoconferencing device. We will briefly cover all three options below, and explain why the third method is preferred.

Approach #1: Using the Anonymous Device

Version 3.0 (a.k.a "Encore") introduced a new option for routing calls to an H.323 network. The feature is known as the "Anonymous Device", and is seen when configuring an H.323 gatekeeper in CallManager via the checkbox labeled "allow anonymous calls". Enabling this checkbox allows CallManager to send Admission Requests (ARQ's) to an H.323 gatekeeper to resolve the location of a dialed number.

However, the use of this feature was specifically designed for inter-cluster calls between CallManager clusters. The reason for this distinction is that CallManager adds some proprietary information to the H.225 setup message when routing calls via the anonymous device. These proprietary additions are referred to as the Cisco inter-cluster trunk protocol. The proprietary nature of this protocol is expected to increase over time, and as a result, it is not recommended that you use this method for routing calls to non-Cisco CallManager devices. However, it is possible that a future version of CallManager could provide a function similar to the anonymous device, which does not use the inter-cluster trunk signaling. If and when that happens, approach number will be become the most preferred approach.

Approach #2: Defining the IP/VC MCU as an H.323 gateway

As of the current release of CallManager, which is the 3.1 (a.k.a "Bravo") release, CallManager uses the well-known H.323 call signaling port of 1720 for all H.323 gateway devices. Unfortunately, the IP/VC 3500 series MCU's and gateways do not use this well-known port number. The MCU's listen on port number 2720, and the gateways listen on port 1820. There is currently no way in CallManager to define what port number to use on a per-gateway basis. It is possible that a future version of CallManager could allow you to do this.

Therefore, defining the IP/VC 3500 series MCU's and gateways as H.323 gateway devices in CallManager is not the recommended method. For the time being, the Cisco MCM Proxy is the best approach, as described below. However, once the ability to specify port numbers for a given H.323 gateway device in CallManager is available, or the IP/VC devices are modified to use the standard port numbers, using the proxy will no longer be the most suitable approach, and approach number two, or a future non-proprietary version of approach number one, will be preferred.

Approach #3: Defining the Cisco MCM Proxy as an H.323 gateway

The Cisco MCM proxy is an H.323 device that registers with the gatekeeper and similar service as the MTP resources in IP telephony environments. It terminates one leg of a call, and establishes the next leg either to a remote proxy, or directly to the destination endpoint being called. This is often used to provide QoS for videoconferencing traffic, because the proxy is able to stamp the packets with an IP precedence marking, or generate an RSVP reservation for the proxied leg of the call. It is also useful for firewall, NAT and security applications.

Fortunately, the proxy is also an excellent tool for our purposes here too. By defining the proxy as an H.323 gateway device in CallManager, and using it for all calls from IP phones to H.323 videoconferencing devices, you gain the following benefits:

1. You do not need to use the anonymous device, thereby steering clear of the proprietary inter-cluster trunk protocol.

2. You do not need to statically define all of your IP/VC devices as H.323 gateways in CallManager.

3. You only have to define one H.323 gateway device (i.e. the proxy), and by way of the H.323 RAS procedure, you can call any H.323 videoconferencing device on the network, whether it is an MCU, gateway or endstation. This is because the proxy will send an ARQ to the gatekeeper to resolve the address for the dialed number. The destination could be any H.323 device in that gatekeeper's zone, or in any remote zone known to the gatekeeper.

Limitations of the Cisco MCM proxy

However, you should be aware of the calls per second and number of sustained calls that the proxy can handle. The number of calls the proxy can handle obviously increases based on the router platform you are using for the MCM proxy software. Table 1 defines these numbers for each router platform. You can see that the numbers are based on video calls at different speeds. Voice calls would be approximately 50% more than a 128kbps call, so for example, the 72xx should be able to do about 150 voice only calls.

Table 1: Number of simultaneous proxied calls per platform

Multimedia Conference Manager MCM

Chassis	IP Routing	H.323 Endpoint Registration	Simultaneous Video Calls	Video Proxy Sessions
72XX	50-100K pps	3000	500	50 @ 768kbps 75 @ 384kbps 100 @ 128kbps
3660	25-100K pps	1800	250	25 @ 768kbps 35 @ 384kbps 50 @ 128kbps
3640	10-40K pps	1800	150	10 @ 768kbps 15 @ 384kbps 30 @ 128kbps
3620	10-15K pps	1800	75	10 @ 768kbps 15 @ 384kbps 30 @ 128kbps
262X	5-10K pps	900	60	2 @ 768kbps 4 @ 384kbps 8 @ 128kbps
261X	2-5K pps	900	60	2 @ 768kbps 4 @ 384kbps 6 @ 128kbps
3810	2-5K pps	900	60	2 @ 768kbps 4 @ 384kbps 6 @ 128kbps
25XX	N/A	600	30	2 @768kbps 4 @ 384kbps 10 @ 128kbps

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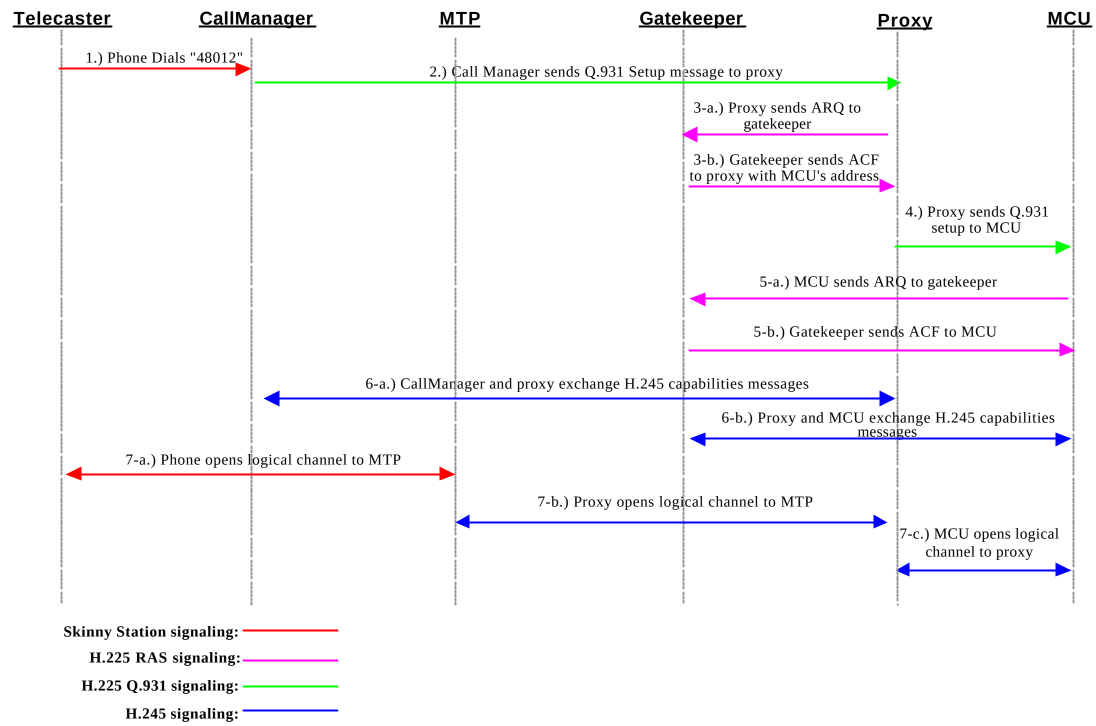
To illustrate how the proxy is used to process the call, consider the following diagram.

Figure 4: Using the proxy for IP phone to H.323 videoconferencing calls.



To accomplish this, there are four simple steps for configuring the CallManager. They are:

4. Configure the proxy as an H.323 gateway device.
5. Configure a route pattern, route list and route group to point to the proxy.
6. Change the default packet size to 30ms.

Configuring the proxy as an H.323 gateway device

To configure the proxy as an H.323 gateway in CallManager, perform the following steps:

1. Click “Device” on the Cisco CallManager Administration page and choose “Add a New Device”.
2. Select “Gateway” as “Device type”.
3. Click “Next”.
4. Select “H.323 Gateway” as “Gateway Type”.
5. Select “H.225” as “Device Protocol”.
6. Click “Next”.
7. Enter the proxy’s IP address or DNS hostname as the “Device Name”.
8. Select a “Device Pool” as appropriate for your network configuration.
9. Check “Media Termination Point Required” if supplementary services are desired.
10. Select a value of “Num Digits” to reflect the length of your IP phone's DN (Directory Number).
11. Check “Sig Digits”.

12. Click “Insert”.

Note: Step 7 can become a problem is the proxy’s IP address is the same as the gatekeeper’s IP address. If you are also using an H.323 gatekeeper for inter-cluster calls, it is recommended that you choose a different interface for the proxy, so that the IP addresses do not conflict with each other. CallManager will not allow you to define the same IP address for both devices. You can use any other available Ethernet or Serial interface for the proxy, or you could create a loopback interface for the proxy.

Note: Step 9 is essential if you want the IP phone to be capable of using the hold and transfer features. Refer to the “Limitations of the IP/VC 3500 series MCU’s” section above for more details on this topic. In addition, the steps above assume that you have already configured device pools, regions, MTP resources, and that you know how many digits to use for the “Num Digits” and “Sig Digits” options. Refer to the CallManager documentation for more details on these options. Finally, the example above was done using version 3.0(8). Other versions may have slightly different fields, but the configuration process and principles are the same.

Configuring route patterns, route lists and route groups to point to the proxy

After configuring the proxy as an H.323 gateway device, you must now configure CallManager to know how to match a string of dialed digits and route the call to this device. The number of route patterns and route groups you need will vary depending on the size and complexity of your network. You may have multiple H.323 MCU’s, gateways and terminal, and multiple paths to those devices in your network. Also, you may have multiple clusters in different area codes and regions, and so you may need multiple route patterns. For example, you may want people in the San Jose cluster to reach the MCU in San Jose via a 5-digit extension, but to reach an MCU in a different city/state, they will have to dial seven or ten digits, etc. Figure 1 shows a very simple topology with only one 3540 MCU and a few H.323 videoconferencing terminals in a single zone, and so we will only have one route pattern, using 5-digit extensions, and a single route group pointing to the proxy.

These items must be configured in the following order:

1. Route group
2. Route list
3. Route pattern

Route group

To configure a route group that includes the newly created H.323 gateway device (i.e. the proxy), perform the following steps:

1. Click “Route Plan” on the Cisco CallManager Administration page and choose “Route Group”.
2. Enter a descriptive name such as “H.323 Proxy” as the “Route Group Name”.
3. Click “Continue”.
4. Select the name of the proxy device you created (whether IP address or DNS name) from the “Device Name” list.
5. Click “Insert”.

Route list

To configure a route list that includes the newly created route group, perform the following steps:

1. Click “Route Plan” on the Cisco CallManager Administration page and choose “Route List”.

2. Enter a descriptive name such as “to Video devices” as the “Route List” name.
3. Click “Insert”.
4. Click “Add Route Group”.
5. Select the name of the route group you defined above, such as “H.323 Proxy” from the “Select Route Group” list.
6. Click “Add”.
7. Click “Insert”.

Route pattern

To configure a route pattern to point to the route list and route group the proxy belongs to, perform the following steps:

1. Click “Route Plan” on the Cisco CallManager Administration page and choose “Route Pattern”.
2. Enter “4xxxx” in the “Route Pattern” field. Therefore a dialed string such as 48123 will match this Route Pattern, where “4” is the last digit of the H.323 zone prefix the MCU is registered in, “8” is the last digit of the MCU’s service prefix and “xxx” would be the unique conference ID the MCU will use to either create a new conference or add the caller to an existing conference.

Note: See the whitepapers listed at the beginning of this paper for more details on the use of service prefixes and conference ID’s on the MCU.

3. Select the name of the route list you defined above, such as “to Video devices” from the “Gateway/Route List” menu.

Note: Optionally, you can skip the configuration step of the route list above, and point the route pattern directly to the H.323 gateway device. This is easier if you only have one MCU and one route pattern. The use of the route list and route groups enables a more complex environment where you may have multiple paths to the MCU, or multiple MCU’s.

4. Check “Route this pattern”.
5. For our example, uncheck the “Provide Outside Dial Tone” option.
6. For our example, leave the “discard digits” field blank. If using a route pattern that contains a prefix that needs to be stripped off, such as XX.XXXXX, the route pattern will discard digits in front of the dot before passing it to the route list. This is optional depending upon the dial plan you decide to use.
7. (Optional) enter 40885 in the “Prefix Digits (Outgoing Calls)” field. By doing this, the CallManager will append 40885 to the number dialed, which in our example is 48123. This will result in the MCU receiving a call destined to 4088548123, where 408854 is the complete zone prefix, “1” is the last digit of the service prefix, and “23” is the unique conference ID.
8. Click “Insert”.

Note: Step 7 is important if you are using a 10-digit dialing format for your H.323 dial plan, but a 5-digit format for your IP phones. If the MCU is using ten digits (actually eight digits with a wildcard at the end for the conference ID, i.e. 40885481) then sending only the last five digits will fail. Therefore, if you are using 10-digit dialing in your H.323 zone and on your MCU, then CallManager must prepend the additional digits to the dial string before signaling to the H.323 proxy. This allows the IP phones to continue using a 5-digit dialing approach to reach the MCU, and CallManager takes care of the digit translation.*

See the whitepapers listed at the beginning of this paper for more details on the use of service prefixes and conference ID’s on the MCU. Specifically, the paper titled “Routing Gateway and

MCU Service Prefixes on the Cisco MCM Gatekeeper" gives a detailed overview of how to configure an H.323 dial plan using the Cisco MCM gatekeeper and the IP/VC MCU's and gateways.

Changing the default packet size to 30ms

As discussed above in the "Limitations of the IP/VC 3500 series MCU's" section, the Cisco IP phones use 20ms as their default packet size. This value is configurable in CallManager, as outlined below. However, it is important to note changing this value is a global change that effects all IP phones registered with that CallManager, so be sure that you want all phone traffic to use 30ms before modifying this value.

To change the default packet size to 30ms, perform the following steps:

1. Click "Service" on the Cisco CallManager Administration page and choose "Service Parameters".
2. Select the CallManager server that you want to modify, and choose "Cisco CallManager" from the list of services.
3. In the "Configured Service Parameters" list, choose "PreferredG711MillisecondPacketSize".
4. Change the value from "20" to "30".
5. Select "Update".

Note: You will need to modify this setting for all CallManagers in the cluster, if you want all IP phones to be able to 30ms.

At this point, you have done all the necessary steps required in CallManager to allow calls *outbound* from an IP phone to the H.323 videoconferencing network. You still must configure the proxy, gatekeeper and MCU(s). We will deal with these items in a later section.

However, to allow calls in the other direction; that is from an H.323 device such as the MCU to an IP phone (which may be useful if you want to "invite" a phone to a conference call, in which the MCU must "dial" the phone and invite it), you must perform some additional configuration steps in CallManager.

Let's look at these steps first, and then we will deal with configuring the proxy, gatekeeper and MCU(s). You can skip this section if you do not require calls to be allowed in both directions.

Configuring the CallManager to register with the gatekeeper

In order for a call to be placed from an H.323 device to an IP phone, the CallManager must register with the gatekeeper. This is because all the H.323 devices, whether they are terminals, MCU's or gateways, use the gatekeeper to perform the address lookup via the H.225 RAS process. Therefore, if the gatekeeper does not know how to route the call to CallManager, calls in this direction will fail.

We will step you through the configuration of both the CallManager and the gatekeeper. However, it is likely that you already have much of these items configured, if your network consists of multiple clusters, and you are using a gatekeeper to perform CAC (Call Admission Control) for inter-cluster calls.

There are two steps required to register the CallManager with the gatekeeper. They are:

- Configure a gatekeeper device in CallManager.

- Configure the technology prefix CallManager will use when registering with the gatekeeper.

Configuring a gatekeeper device

To configure a gatekeeper device in CallManager, perform the following steps:

1. Click "Device" on the Cisco CallManager Administration page and choose "Add a New Device".
2. Select "Gatekeeper" as "Device type".
3. Click "Next".
4. Enter the gatekeepers IP address or DNS hostname as the "Device Name".

Note: The gatekeepers IP address cannot be the same as the proxy's IP address. CallManager will not allow you to define the same IP address for both devices. You can use any other available Ethernet or Serial interface for the proxy, or you could create a loopback interface for the proxy.

5. (Optional) Enter a description for this gatekeeper.
6. Leave the registration timers at their default values.
7. Select "Gateway" from the "Terminal Type" menu.
8. Select a "Device Pool" as appropriate for your network configuration.

Note: The remaining steps are only relevant for inter-cluster calls between CallManagers, which use the gatekeeper for Call Admission Control.

9. (Optional) Select "Allow Anonymous Calls" if you are using this gatekeeper to perform inter-cluster calls between CallManagers. If you are not, leave this option unchecked.
10. Select the appropriate values for "Calling Search Space" and "Location" as appropriate for you network configuration.
11. (Optional) Enter a value for the "Caller ID DN" as appropriate.
12. Choose "Originator" from the "Calling Party Selection" menu.
13. Choose "None" from the "Presentation Bit" menu.
14. (Optional) Select "Media Termination Point Required" if you want to use MTP for calls between CallManager clusters. This option does not dictate the setting for calls from IP phones to the MCU, since you are not using the gatekeeper to route calls in that direction. Recall that your route pattern points to the H.323 gateway device, not to the gatekeeper. Selecting MTP here will only govern calls whose route pattern points to the gatekeeper/ anonymous device (i.e. inter-cluster calls).
15. Select the appropriate values for the "Num Digits", "Sig Digits" and "Prefix DN" options, as appropriate to your inter-cluster dialing plan and network configuration. These options do not govern calls from IP phones to the MCU, as these calls do not use the anonymous device.

Configuring a technology prefix in CallManager

To configure the CallManager to register a technology prefix with the gatekeeper, perform the following steps:

Note: This step is necessary to successfully route calls from an H.323 device to an IP phone. The gatekeeper must know which CallManager to route the call to, and it does this via the technology prefix.

1. Click "Service" on the Cisco CallManager Administration page and choose "Service Parameters".

2. Select the CallManager server that you want to modify, and choose “Cisco CallManager” from the list of services.
3. In the “Configured Service Parameters” list, choose “GatekeeperSupportedPrefix”.
4. From the “Type” menu list, select “String”.
5. In the “Value” field, enter a technology prefix that will be unique in the H.323 network, such as “1#” for example. This value could be anything you want it to be, as long as it does not conflict with any zone prefixes and other technology prefixes/service prefixes already defined in the H.323 network. 1# is the example used in the CallManager documentation as well, so it is the recommended value.
6. Choose “Update”.
7. (Optional) Choose “Reset” to reset the gatekeeper device to force CallManager to register with the gatekeeper immediately. However, there are still some configuration steps to make on the gatekeeper, so you can come back and do this later if you want.

Configuring the gatekeeper

The gatekeeper is the brains of an H.323 network, and facilities the call processing function for H.323 devices, just as CallManager provides the call processing function in an IP telephony network. Therefore, in order for an H.323 device to call an IP phone, the gatekeeper must know the physical location of the CallManager (i.e. its IP address and TCP signaling port information), and it must also have a route plan in place to detect that a string of dialed digits should be routed to the CallManager.

Gatekeepers are virtually guaranteed to exist if the customer already has an H.323 videoconferencing network. In addition, if they are using the Cisco IP/VC products, chances are also high that they will be using the Cisco MCM (Multimedia Conference Manager) as their H.323 gatekeeper.

Likewise, if the customer already has a large, multi-site IP telephony deployment, using the "Distributed Call Processing" architecture model defined in the IP Telephony Design Guide, they most likely are already using the Cisco MCM Gatekeeper to provide CAC for inter-cluster IP telephony calls.

Therefore, depending on the current state of the gatekeeper configuration, you may already have the majority of the following steps in place already. However, we will look at how to configure the gatekeeper from the beginning, using the simple network example giving in Figure 1.

There are four simple steps for configuring the gatekeeper. They are:

7. Configure a local zone for the CallManager.
8. Configure a local zone for the H.323 videoconferencing devices.
9. Configure zone prefixes for both local zones.
10. Configure a technology prefix for the CallManager.
11. Configure hopoff statements for any H.323 technology prefixes (i.e. MCU's and gateways) you have.

Configuring different zones for the CallManager and the H.323 videoconferencing devices

There are many ways that one could configure the gatekeeper to route calls from the MCU to the CallManager, but as the "best practice" approach, we recommend using different zones for these two types of devices. There are several benefits for doing so. Two of those benefits are:

- Allows for a more scalable and flexible dial plan between your voice and video devices.
- Allows for more granular bandwidth controls, especially when using the gatekeeper for CAC between CallManager clusters.

Configuring the CallManager zone

To configure a zone in the gatekeeper for the CallManager to register with, enter the the following commands at the gatekeeper configuration prompt:

1. "zone local CM cisco.com 10.1.1.1"

   Where "CM" is the name of the zone, "cisco.com" is the domain and "10.1.1.1" is the instance of the IP address of the gatekeeper (A router may have multiple IP interfaces/addresses. You need to specify which address the gatekeeper will use).

2. "zone prefix CM 3…."

   Or alternatively, "zone prefix CM 408853…." depending on whether you want the H.323 devices to have to dial all ten digits to reach the IP phone or just a 5-digit extension. This choice will be determined by the complexity of your H.323 dial plan. Please refer to the Cisco Videoconferencing Design Guide for more details on choosing zone prefixes.

3. “zone subnet CM <IP subnet of CallManager/ subnet mask> enable”
4. “no zone subnet CM default enable”

   The combination of commands 3 and 4 forces the CallManager to register in the correct zone. Although new versions of CallManager may allow you to specify the zone name you want to register with, (i.e. the gk-id) most videoconferencing devices, including the IP/VC products do not. Therefore, when using one gatekeeper for voice and video zones, you must use the zone subnet commands to force the devices onto the correct zone.

5. “gw-type-prefix 1# default-technology”

   This tells the gatekeeper to use 1# as the default technology prefix if it does not find a match for the dialed number elsewhere. Since CallManager is registering with this technology prefix (configured above), the gatekeeper will route the call to that CallManager’s IP address.

Note: It is important to understand the way the zone prefix and the default technology prefix work. If an H.323 device dialed 31234, the Gatekeeper will find a match against the local zone prefix 3…. for the “CM” zone. It will then look to see if there is an endpoint in that zone that has registered the E.164 address 31234. It will not find one, and so will use the default technology prefix to forward the call to any device that has registered with that technology prefix, which in this case, is the CallManager.

Depending on the size and complexity of your IP telephony network, you may have several CallManagers registering with the gatekeeper, and each of them may be registering with the technology prefix 1#. You would then use separate zones (like one for 3…., one for 4…. and so on) to ensure that the call gets routed to the appropriate CallManager cluster.

Configuring the videoconferencing zone

To configure a zone in the gatekeeper for the H.323 videoconferencing devices to register to, enter the the following commands at the gatekeeper configuration prompt:

1. “zone local VIDEO cisco.com 10.1.1.1”
2. “zone prefix VIDEO 4*”

   Or alternatively, “zone prefix VIDEO 408854*”, depending on whether you want the H.323 device to have to dial all ten digits to reach other H.323 devices in the same zone or just a 5-digit extension. This choice will be determined by the complexity of your existing H.323 dial plan.

Note: You may notice that we use an asterisk (*) when defining the video zone prefix, but we used dots (…) when defining the CallManager zone prefix. The reason for this is that IP/VC MCU’s and gateways register with an E.164 address that is made up of a randomly generated number. This number is typically 13 digits long, and so using dots as wildcard digits for the zone prefix would result in the gatekeeper rejecting the MCU’s and gateways registration requests. This is currently filed as a bug, and may be fixed in a future version of the IP/VC products. Until then, use an asterisk instead of dots for the zone prefix wildcard digits.

3. “zone subnet VIDEO <IP subnet of H.323 video devices / subnet mask> enable”
4. “no zone subnet VIDEO default enable”

   The combination of commands 3 and 4 forces the H.323 videoconferencing devices to register in the correct zone. Today, most videoconferencing devices, including the IP/VC products do not allow you to specify the name of the zone in the registration requests (i.e. the gk-id). Therefore, when using one gatekeeper for voice and video zones, you must use the zone subnet commands to force the devices into the correct zone.

5. “no shutdown”.
6. (Optional) “no use-proxy VIDEO remote-zone CM inbound-to terminal”
7. (Optional) “no use-proxy VIDEO remote-zone CM outbound-from terminal”

Note: Steps 6 and 7 depend on whether you want to use the H.323 proxy for inter-zone H.323 calls. The default behavior of the MCM for a newly defined zone is as follows:

```
PROXY USAGE CONFIGURATION :
  Inbound Calls from all other zones :
    to terminals in local zone VIDEO : use proxy
    to gateways in local zone VIDEO  : do not use proxy
    to mcu's in local zone VIDEO  : do not use proxy
  Outbound Calls to all other zones :
    from terminals in local zone VIDEO : use proxy
    from gateways in local zone VIDEO  : do not use proxy
    from mcu's in local zone VIDEO  : do not use proxy
```

We strongly urge you to read the Multimedia Conference Manager documentation for more information on this, but simply put, you should carefully consider whether you want to use the proxy for all IP phone to H.323 video calls, and visa-versa. This decision will also depend on which of the three approaches you use to route calls from CallManager to the H.323 videoconferencing devices. For example, using approach number one, the CallManager will query

8. “gw-type-prefix 48* hopoff VIDEO” (Optional)

Note: This command is needed when the CM zone and the VIDEO zone are on the same gatekeeper (i.e. two local zones, as opposed to using two gatekeepers, and configuring one local zone per gatekeeper, and pointing to the other zone as a “remote zone”. This is due to the way the gatekeeper's parsing logic works. When you dial “48012”, the gatekeeper will match the technology prefix of the MCU, which is “480”, then it will attempt to match the remaining digits (“12”) against any known zone prefixes. It will fail to find a match for a zone prefix, since the CM zone is “3…” and the VIDEO zone is “4”, and will by default then use the source zone to look for a matching E.164 address. It will not find an E.164 address in the CM zone of “12”, and will then use the default technology prefix (1#), and route the call back to CallManager, which of course will not work. You can display the gatekeeper's parsing logic by doing “debug gatekeeper main 10” when placing the call.*

The alternative to using this hopoff statement is to use configure the MCU to register as an “MCU” terminal type, rather than as an “H320-GW” terminal type, which is the factory default for the MCU. Registering as an MCU will cause the MCU to register its services as E.164 addresses, rather than as technology prefixes. This will allow the call to be routed between zones properly, but you will loose the ability to add additional digits to the dial string to denote a conference ID. Therefore, you would have to change the service prefixes in your MCU from “480” to “48012”, etc.*

For more information on the gatekeeper parsing logic, and the differences between using technology prefixes versus E.164 addresses on the MCU, please see the whitepapers entitled “Routing MCU and Gateway Service Prefixes on the Cisco MCM Gatekeeper”, and “Configuring Services on the IP/VC 3500 Series MCU's and Gateways”. Again, this problem is only evident when using local zones on one gatekeeper. If you use separate gatekeepers for your CM and VIDEO zones, this parsing order problem does not occur.

Configuring the MCU

You must now configure the MCU to register with the gatekeeper, and configure the appropriate services and service prefixes on the MCU to host the types of conferences that you wish to use. As described above in the “Capabilities of the IP/VC 3500 series MCU’s” section, the MCU is capable of a variety of different calls speeds and video formats, as well as audio-only or audio/ video on a per service basis.

Configuring the MCU to register with the gatekeeper

To configure the MCU to register with the gatekeeper, perform the following steps:

1. For the 3540 model, logon to the MCU’s Administration page by entering “http://<MCU’s IP address>/admin/mcu/default.asp” in your browsers address field.
2. On the “Basics” tab, enter the IP address of the gatekeeper in the “Gatekeeper IP” field.
3. For the 3510 model, use the Configuration Utility application to logon to the MCU, click “Unit Setup”, and then click “Next” twice. This will bring you to the miscellaneous tab, where you can define the gatekeeper’s IP address.

Note: You must reset the MCU before it will register with the gatekeeper. However, you will need to do so again after configuring the services below, so you can wait and do it all at one time after completing the following steps.

Configuring a service to support audio-only calls

To configure a service to support audio-only calls, perform the following steps:

1. For the 3540 model, click the “Services” tab from the MCU’s Administration page.
2. For the 3510 model, use the Configuration Utility application to logon to the MCU, click “Unit Setup”, and then click “Next” once. This will bring you to the service definition table.
3. Click “Add”.
4. Enter a service prefix number in the “Prefix” field. This number should match your H.323 dial plan, and should be unique. This is the number that users will dial to access this MCU’s service, along with a unique conference ID. For our example network topology and dial plan shown in figure 1, enter 40885480

Note: Notice that the service prefix is only eight digits, and there are no dots (..) or asterisk (*) wildcard digits at the end. When the MCU registers with the gatekeeper, the gatekeeper will assume a “” wildcard at the end of the number, and route any calls that match the first eight digits to the MCU. The MCU will allow users to add additional digits for the conference ID to the end of the service prefix, such as 4088548012, where “12” is the conference ID chosen by the initiator of the conference. If using the MCU in scheduled-only mode, the scheduling software will generate a random conference ID for the user to use for that conference instance.*

5. Enter a description of the conference service in the “Description” field.
6. Select “Voice Only”.
7. Enter a numeric value in the “Reserved Number of Parties” field. This is how many users will be allowed to join the conference, unless the “Allow Dynamic Expansion” option is also selected.

Note: Dynamic expansion is not allowed if the MCU is configured for “scheduled only” mode. Dynamic expansion is only allowed when using the MCU in “ad hoc” mode. See the MCU’s Administrators Guide and Release Notes for more information on this topic.

8. Click “OK” when finished.

Note: You must "upload" the changes and reset the MCU before the changes will take effect. If you are only going to enable one service, you may perform the reboot now. However, you will need to do so again after configuring the video service below, so you can wait and do it all at one time after completing the following steps.

Configuring a service to support combined audio and video calls

To configure a service to support audio and video calls, perform the following steps:

1. For the 3540 model, click the "Services" tab from the MCU's Administration page.
2. For the 3510 model, use the Configuration Utility application to logon to the MCU, click "Unit Setup", and then click "Next" once. This will bring you to the service definition table.
3. Click "Add".
4. Enter a service prefix number in the "Prefix" field. This number should match your H.323 dial plan, and should be unique. This is the number that users will dial to access this MCU's service, along with a unique conference ID. For our example network topology and dial plan shown in Figure 1, enter 40885481 (recall that you used 0 as the last digit for the voice-only service…).
5. Enter a description of the conference service in the "Description" field.
6. Select "Voice + Video".
7. Decide which video format you want to use; "Voice-Activated" or "Continuous-Presence", and select the appropriate option from the "Continous-Presence" menu ("disabled" = voice-activated).
8. Depending upon your choice for step 7, choose a video bit-rate from the "Incoming Bit-Rate" menu. For example, if you want to use 384kbps in continuous-presence mode, you would enter "80" in the incoming bit-rate field *(80kbps per participant, times four quadrants = 320kbps, plus 64kbps for G.711 audio = 384kbps).*
9. Select the appropriate settings from the "Video Format", "Picture Format" and "Frame Rate" options. These options will vary depending upon your choice for step 7.
10. Enter a numeric value in the "Reserved Number of Parties" field. This value dictates how many users will be allowed to join the conference, unless the "Allow Dynamic Expansion" option is also selected.
11. Click "OK" when finished.

Uploading the new parameters to the MCU and reboot it

Now that you have configured the MCU to register with the gatekeeper, and have configured one or more services on the MCU, you must now upload your changes, causing the MCU to reset. To do so, on the 3540, simply click on the "Upload" button, and respond to the dialog box to confirm your changes. On the 3510, click next until you are prompted to "upload the parameters to the unit".

Configuring gateways

So far we have discussed how to configure routing from an IP telephony network of Cisco IP phones to an H.323 videoconferencing network, and visa versa. The last point that needs to be considered is how to bring PSTN devices into the network via gateways.

Cisco offers an incredible array of gateways to meet the different needs of customers. These gateways offer a variety of PSTN interfaces, such as FXS, FXO, E&M, T-1 CAS, BRI and PRI. In addition, these gateways also offer a variety of signaling methods, such as H.323, MGCP or CSSP (Cisco Skinny Station Protocol).

All of these gateway models support audio, but currently only the IP/VC 3500 series gateways support video in addition to audio. In other words, the IP/VC gateways support audio codecs such as G.711, G.728, G.722, as well as video codecs such as H.261 and H.263.

At this point you may ask, *"why do I need to buy one of the audio-only gateways then, if I can just buy an IP/VC gateway and get both audio and video support through it"?*

This is a valid question, and technically you could certainly use the IP/VC gateway, configure it in CallManager, and use it for your telephony calls as well. However, you should carefully evaluate the capabilities of the IP/VC gateways in comparison to the other Cisco gateway models when making this decision. There are several key benefits to using some of the other Cisco gateways, including, but not limited to, the following:

- The Cisco gateways offer a much wider selection of interfaces, and a much larger amount of port density.
- The Cisco gateways run Cisco IOS Software and/or run in Cisco Catalyst switches, making them much more reliable, manageable and scalable.
- The Cisco gateways offer MGCP and CSSP support. MGCP in particular, when paired with the 3.1 release of CallManager, offers nearly 100% call survivability. If the primary CallManager goes down, the calls will survive. This is not true of H.323 gateways, and the IP/VC gateways only support H.323 at this time.

Note: For more information on call survivability for voice gateways in CallManager, please see the Whitepaper referenced at the beginning of this document.

- The Cisco IP/VC gateways do not currently support G.723 or G.729 audio codecs, which are popular codecs for VoIP applications. Rather, they only support codecs most typically used in videoconferencing applications, specifically G.711, G.728 and G.722
- The Cisco IP/VC gateways lack the load-balancing and redundancy capabilities of the Cisco IOS and Catalyst based gateways. The Cisco gateways offer redundancy through a variety of options, such as through redundant, prioritized dial-peers, or through MGCP signaling using CallManager's capabilities to provide redundant routes. For example, the Cisco IP/VC gateways have no ability to "hair-pin" calls back out to the PSTN in the event the IP network is congested.

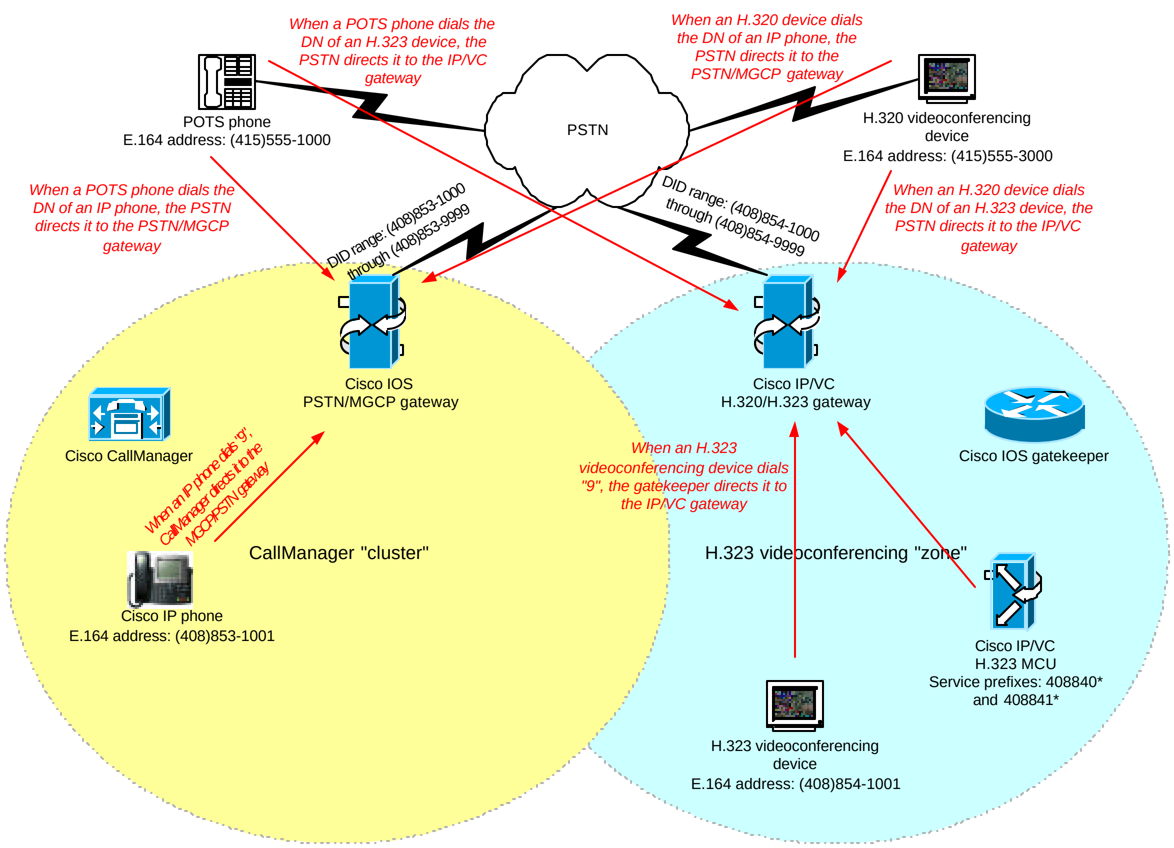
In light of the above evidence, we believe that the vast majority of customers will choose to purchase and deploy separate gateways for voice and videoconferencing. Whether the voice gateways operate in MGCP, H.323 or SCCP mode - with MGCP being the most common due to its additional management and call survivability characteristics – these gateways will all be controlled by CallManager in an enterprise environment, or by the new SRST (Survivable Remote Site Telephony) service in the event of a WAN failure at a branch office. The video gateways, by contrast, will all be controlled by the gatekeeper, at least for the foreseeable future, until the need for a gatekeeper is superimposed, perhaps by a new version of CallManager that supports video devices and negates the need for a gatekeeper, or by some other fundamental change to the product offerings available today.

Given this fact, referring back to Figure 1, you see that a typical network may, and most likely will, consist of a variety of gateways. Whatever the case, it is safe to say that PSTN callers, whether they are POTS phones, cellular phones, or H.320 videoconferencing devices, can access any of the devices on the enterprise network, whether they are IP phones or H.323 videoconferencing devices, via one these gateways. Likewise, any of the devices on the enterprise IP network, whether they are IP phones or H.323 videoconferencing devices, can access the PSTN via these same gateways. Which gateway is used depends on what call processing entity the device is registered with.

For example, dialing "9" on an IP phone will give you PSTN access via the closest voice gateway, whereas dialing "9" on a video device will send you to a video gateway that has registered the prefix 9* with the gatekeeper. For inbound calls, the gateway used depends on the DN of the device. You most likely will have a range of DID numbers for your IP phone, and those DN's are associated to a particular T-1 or PRI circuit, which of course terminates into a particular gateway. Likewise, your H.323 videoconferencing devices will need either a DID number, or some other E.164 address accessible from the PSTN via IVR or TCS4, and those DN's would be associated with the PRI or BRI circuits of the IP/VC gateways.

We will not be providing any specific configuration steps for the gateways in this paper. Please refer to the appropriate users guides for that information. However, the following diagram illustrates how various gateways would offer connectivity to/from the PSTN to/from IP phones or H.323 videoconferencing devices, and how dialing "9" from an IP phone would differ from dialing "9" on a H.323 videoconferencing device.

Figure 5: Seamless gateway connectivity to/from IP phones and H.323 videoconferencing devices



Bringing it all together: a summary of the last few sections

Now that you have configured the CallManager (with MTP enabled), the gatekeeper, the MCU and any gateways you might have, you should be ready to establish calls in any direction.

The following sections will help clarify for you the overall design approach used in the above sections, explain how the call signaling will work, and demonstrate some example call scenarios.

The biggest thing to point out, again, is that the use of the "Anonymous Device" in CallManager for calling anything other than another CallManager cluster is currently not recommended (see the explanations in the "Configuring CallManager" section on page 11). Therefore, rather than reaching the IP/VC videoconferencing devices via the gatekeeper/anonymous device, we have statically defined the MCM proxy in CallManager as an "H.323 Gateway". As a result, for calls from an IP phone to the MCU, or to any other H.323 videoconferencing device, the CallManager will signal directly to the proxy via H.225 signaling. The proxy will then send an ARQ (Admission Request) to the gatekeeper to resolve the dialed number.

By using the proxy in this fashion, you also get the added benefit that, if you have a hundred other H.323 videoconferencing devices, each of them does not have to be statically defined in CallManager. The proxy will resolve any address dialed via the gatekeeper RAS process. However, you should be aware of the calls per second and number of sustained calls that the proxy can handle. See the section above entitled "Limitations of the Cisco MCM Proxy" for more details on this issue.

In the opposite direction, the MCU uses the gatekeeper to perform all address resolution. Therefore, a call from the MCU to an IP phone will go through the standard RAS process; the gatekeeper will match the dialed string to the default technology prefix of the CallManager for that zone, and inform the MCU of the CallManagers IP address and TCP signaling port information. The MCU will then begin signaling the CallManager directly with H.225 messages, and initiate the call establishment sequence.

Finally, connectivity to/from PSTN users is available through a variety of gateway options. Furthermore, due to the availability of MGCP in the Cisco gateways, and the increased call survivability it affords, the IP/VC 3500 series gateways are not recommended for voice-only applications. Customers should use the Cisco IOS-based gateways and/or the Catalyst line card-based gateways for their voice calls, and the IP/VC 3500 series gateways for video calls. In other words, we do not recommend using an IP/VC gateway instead of a voice gateway for calls between your IP phones and the PSTN. You may however want to use a voice gateway for calls between PSTN audio-only devices and your IP/VC MCU's for example. This is perfectly acceptable, and in fact can provide a very nice audio-only conferencing solution if designed properly.

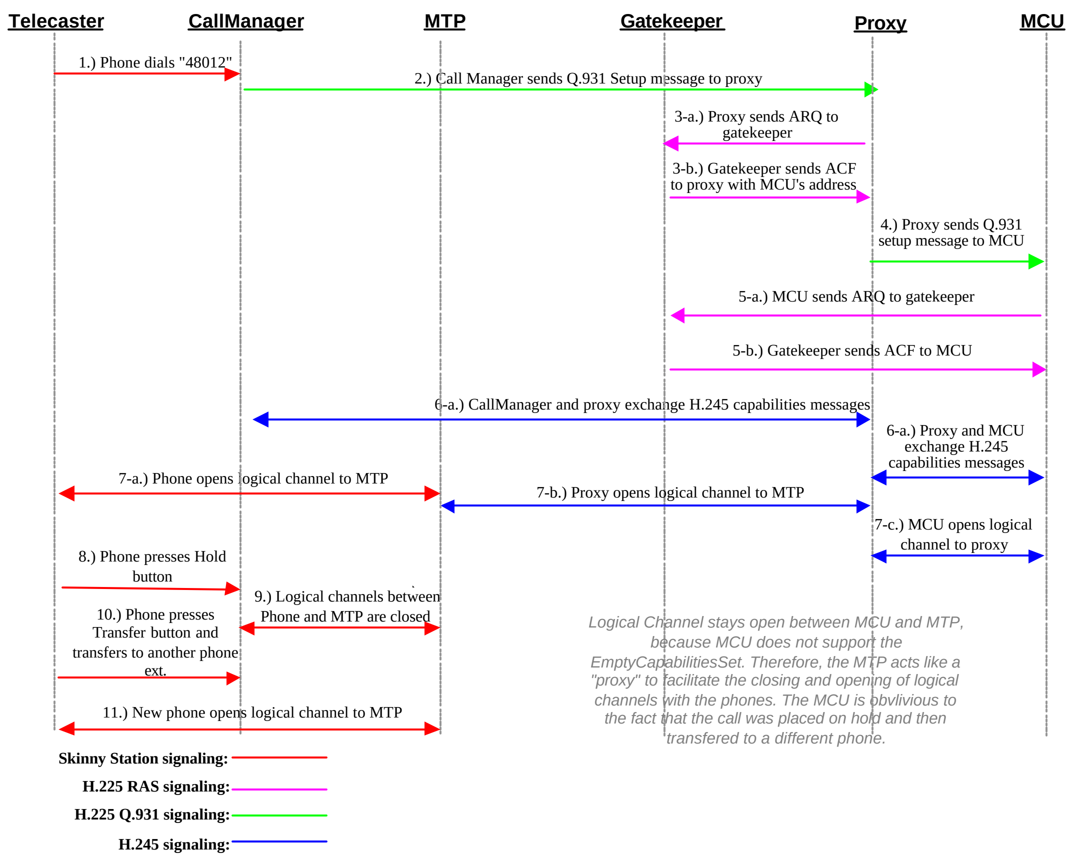
The following section will present several example call scenarios to better illustrate how the signaling and call routing will work. Examples include IP phone to H.323 video, H.323 video to IP phone, POTS phone to H.323 video, H.323 video to POTS phone, etc.

Example call scenarios

Example 1: IP phone to MCU conference with hold and transfer functions

In this example, you can see that the proxy is used by CallManager to route the call to the H.323 videoconferencing network, and the proxy takes care of resolving the number dialed to the MCU's IP address by way of the RAS process (step 3). The RTP streams also flow through the proxy in this example. In addition, the MTP is also in the RTP path, because the MCU does not support empty capabilities.

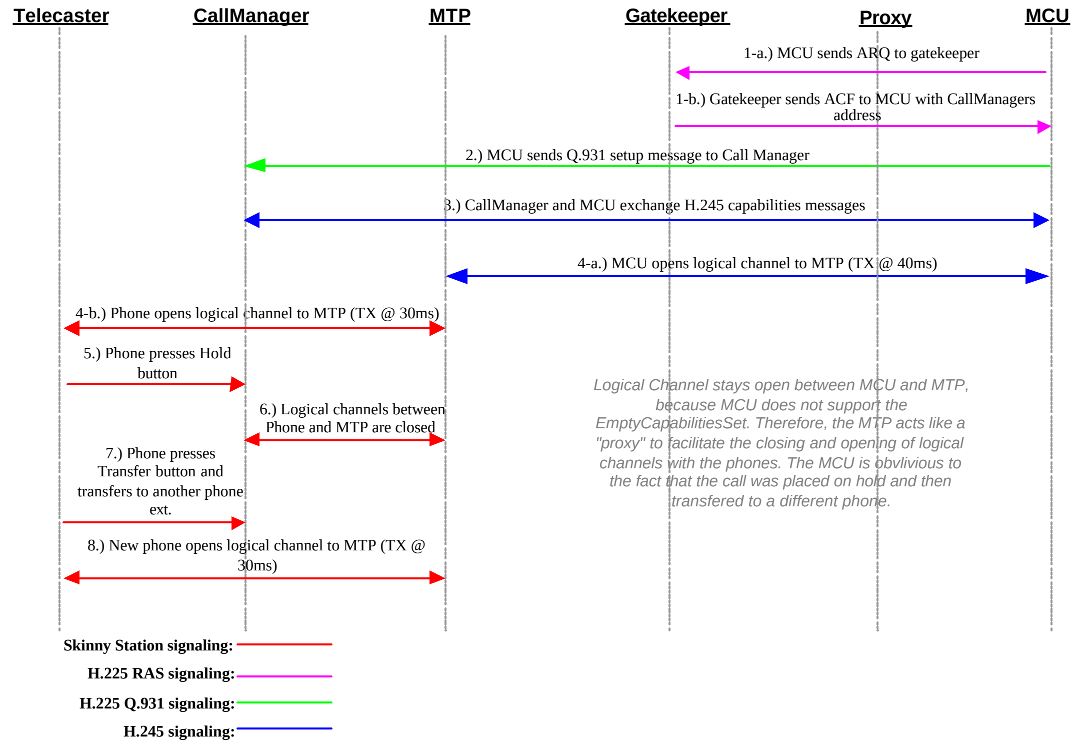
Figure 6: Call signaling from IP phone to MCU conference with hold and transfer



Example 2: MCU to IP phone with hold and transfer functions

In this example, you can see the proxy is not used, because we have `“no use-proxy VIDEO remote-zone CM outbound-from mcu”` configured. The MCU resolves the number of the IP phone via the gatekeeper RAS process, and the MCU and the CallManager talk H.245 directly. However, the MTP is still required because the MCU does not support empty capabilities.

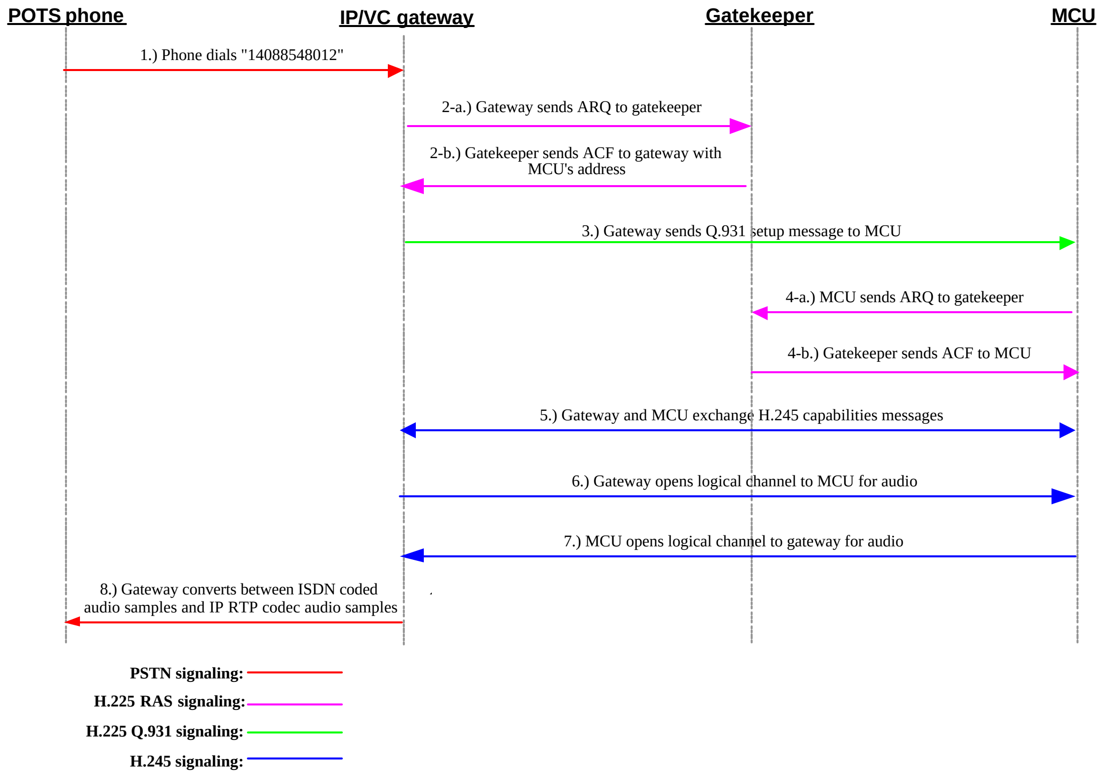
Figure 7: MCU to IP phone with hold and transfer



Example 3: POTS phone to MCU via IP/VC gateway

In this example, the gateway resolves the number dialed by way of the gatekeeper RAS process, and connects the call into the MCU conference. Hold and transfer capabilities are not supported in this scenario, because the IP/VC gateways do not support “hook flash”, and even if and when they do, the MCU does not support empty capabilities. In addition, since the POTS phone is only capable of audio, the gateway only advertises its audio capabilities in its H.245 capabilities exchange with the MCU.

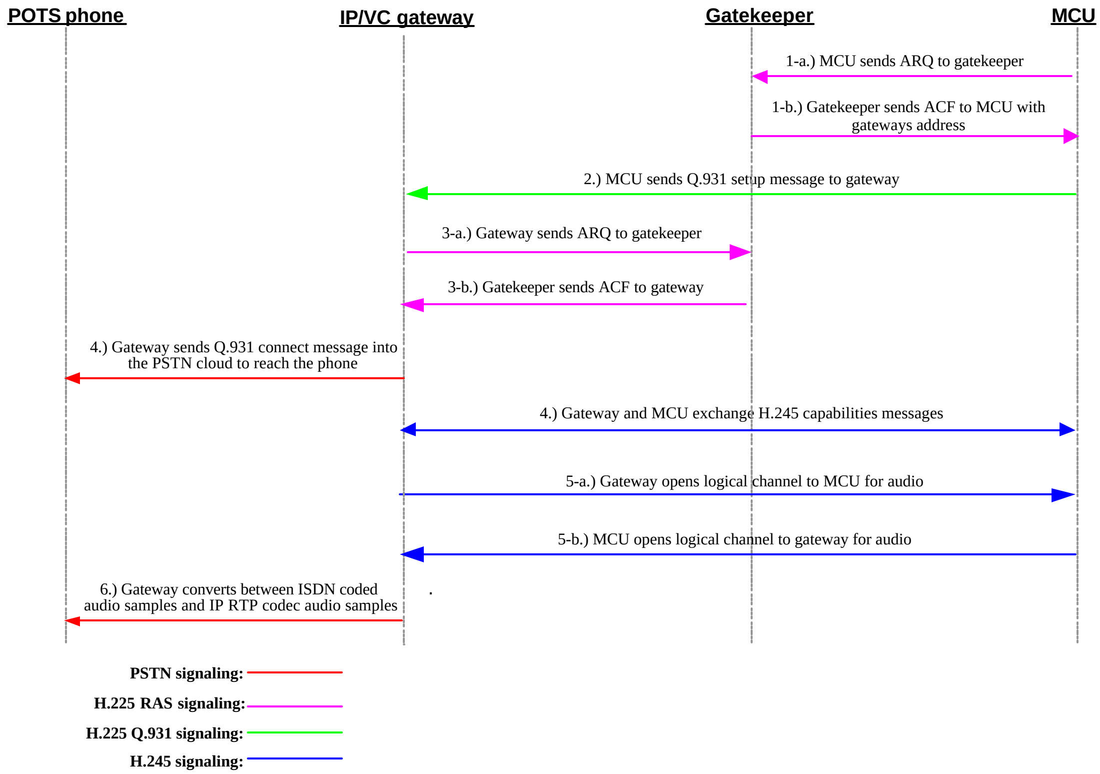
Figure 8: Call signaling example of POTS phone to MCU via IP/VC gateway



Example 4: MCU to POTS phone via IP/VC gateway

In this example, the MCU must dial a gateway access code (i.e. “9”), and the number dialed is resolved by the gatekeeper RAS process, which forwards all calls that start with “9” to the gateway. Hold and transfer capabilities are not supported in this scenario, because the IP/VC gateways do not support “hook flash”, and even if and when they do, the MCU does not support empty capabilities. In addition, since the POTS phone is only capable of audio, the gateway only advertises its audio capabilities in its H.245 capabilities exchange with the MCU.

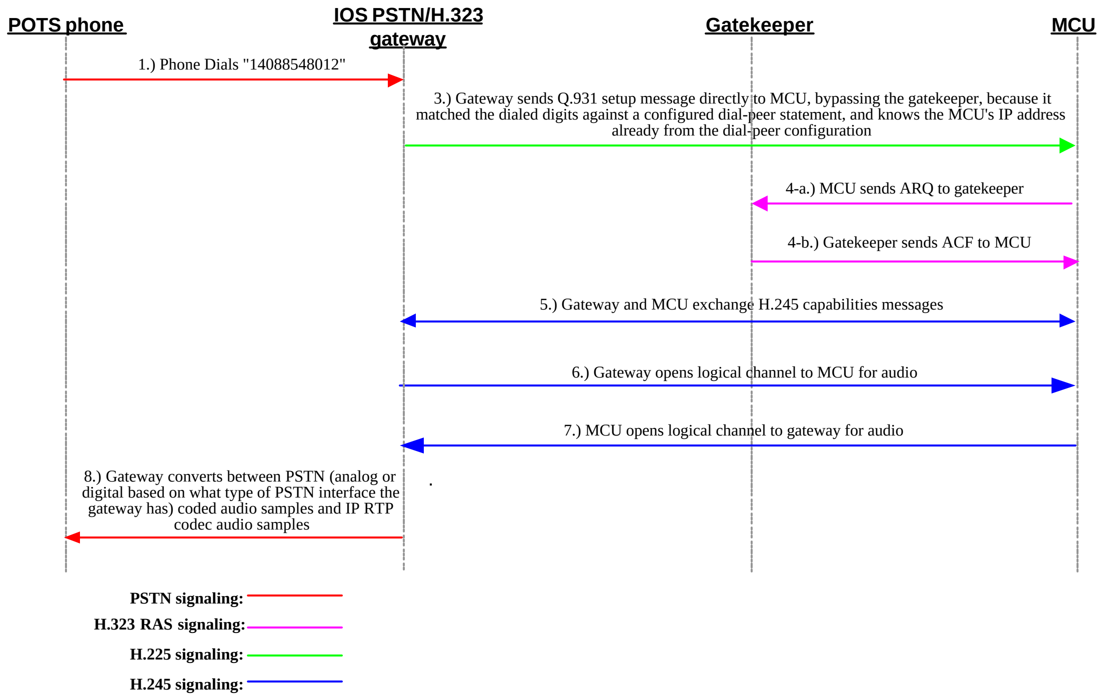
Figure 9: Call signaling example of MCU to POTS phone via IP/VC gateway



Example 5: POTS phone to MCU via IOS H.323 gateway

The Cisco IOS H.323 gateways can use the gatekeeper RAS process to resolve the dialed number to the MCU's IP address, or you can statically configure a "dial peer" in the gateway to route calls matching a certain number pattern directly to the MCU. Since previous examples have demonstrated the gatekeeper RAS process sufficiently, the below diagram illustrates a call using a dial peer statement rather than RAS. Hold and transfer capabilities are not supported in this scenario, because some of the Cisco IOS gateways do not support "hook flash", and even if and when they do, the MCU does not support empty capabilities. In addition, since the POTS phone and the IOS gateways are only capable of audio, the gateway only advertises its audio capabilities in its H.245 capabilities exchange with the MCU.

Figure 10: Call signaling example of POTS phone to MCU via Cisco IOS H.323 gatekeeper using dial-peers



We could give many other examples, but there is probably no need to belabor the point. Outbound calls from an IP phone to the PSTN, or outbound calls from an H.323 videoconferencing device to an H.320 device would work as advertised in the various users guides as well.

For more information on the subject discussed in this paper, if you are a customer, please visit the Cisco Networking Professional Connection at http://www.cisco.com/go/netpro.
If you are a Cisco employee, please contact the AVVID Technical Marketing team, or myself directly at kevin@cisco.com

